ABSTRACT

An input signal enters a noise suppression system in a time domain and is converted to a frequency domain. The noise suppression system then estimates a signal to noise ratio of the frequency domain signal. Next, a signal gain is calculated based on the estimated signal to noise ratio and a voicing parameter. The voicing parameter may be determined based on the frequency domain signal or may be determined based on a signal ahead of the frequency domain signal with respect to time. In that event, the voicing parameter is fed back to the noise suppression system, for example, by a speech coder, to calculate the signal gain. After calculating the gain, the noise suppression system modifies the signal using the calculated gain to enhance the signal quality. The modified signal may further be converted from the frequency domain back to the time domain for speech coding.